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IN REPLY REFER TO: Joint Interoperability Test Command (JTE) 29 Sep 2016

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Extension of the Joint Interoperability Certification of the Cisco Enterprise Session Controller (ESC) 15 Release 10.5.2 System Update (SU) 3

References: (a) Department of Defense Instruction 8100.04, "DoD Unified Capabilities (UC)," 9 December 2010
(b) Office of the Department of Defense Chief Information Officer, "Department of Defense Unified Capabilities Requirements 2013, Errata 1," 1 July 2013
(c) through (h), see Enclosure

1. **Certification Authority.** Reference (a) establishes the Joint Interoperability Test Command (JITC) as the Joint Interoperability Certification Authority for UC products, Reference (b).

2. **Conditions of Certification.** The Cisco ESC 15 Release 10.5.2 SU3; hereinafter referred to as the System Under Test (SUT), meets the critical requirements of the Unified Capabilities Requirements (UCR), Reference (b), and is certified for joint use as an ESC in Type 1, 2, and 3 environments and as a Local Session Controller (LSC) with the conditions described in Table 1. This certification expires upon changes that affect interoperability, but no later than three years from the expiration date of the UC Approved Products List (APL) memorandum.

The extension of this certification is for Desktop Review (DTR) 13, which was requested to update the SX10, SX20, and SX80 video teleconference codec software, as well as update the MX200G2, MX300G2, MX700, MX800 Dual, and MX800 codec software from version Collaboration Endpoint (CE) 8.1.0 to version CE 8.2. See Paragraph 4 for the test details.

Table 1. Conditions

Condition	Operational Impact	Remarks
UCR Waivers		
None.		
Conditions of Fielding		
The SUT does not receive BPA on analog EIs Voice Gateway, line side-Media Gateway, or trunk side-Media Gateway when calling any EI directly off of the SUT that are busy with Equal or Higher Precedence above ROUTINE. To mitigate this condition, the SUT must be configured to divert all calls upon a BPA condition to the alternate directory number in lieu of an announcement.	Minor	See note 1.
To avoid a video interoperability anomaly with the Avaya AS5300 soft client, the AES-GCM Authenticated Encryption in SRTP must be deleted in the SDP OFFER. The methodology for deleting AES-GCM encryption from the SDP OFFER is provided in the SUT DG.	Minor	See note 1.

Table 1. Conditions (continued)

Condition	Operational Impact	Remarks
Open Test Discrepancies		
Per the vendor's LoC, the SUT fails immediately to divert all precedence above ROUTINE calls placed to Jabber ROEIs. The SUT diverts only when the ROEI is busy, if it is idle it will offer the call and divert if not answered.	Minor	See note 2.
Per the vendor's LoC, the SUT video conferencing system does not support the required G.728 audio codecs.	Minor	See note 2.
Per the vendor's LoC, the SUT Media Gateways do not support all required codecs for multiple codecs for a given session. The vendor's LoC states that G.723.1 is not supported for multiple codecs for a given session.	Minor	See note 2.
The SUT does not support Local RTS Database (LRDB).	Minor	See note 2.
The SUT does not support Master RTS Database (MRDB).	Minor	See note 2.
Per the vendor's LoC, the SUT does not support NTPv3.	Minor	See note 1.
Per the vendor's LoC, the SUT is unable to preconfigure OCSP responder based on the issued directory number or provide a preference of one Authority Information Access (AIA) extension over another.	Minor	See note 3.
The SUT DX series VVoIP devices fail to establish two-way video when calls are placed to a Polycom Group Series video EI. This discrepancy was fixed and successfully tested with the update of the IOS IWG/SBC routers from 15.4(3)M5 to 15.5(3)M3 included with DTRs 7 and 11.	Minor	CLOSED
Per the vendor's LoC, the SUT does not comply with Native Session to Modem-Based Session Transition Procedures. This discrepancy was fixed and successfully tested with DTRs 7 and 11, which included the following updates: update UCM from version 10.5.2.12901-1 to version 10.5.2 SU3, and update the 35xx series analog voice gateways, and the 29xx series and 39xx series ISR voice gateways from IOS 15.4(3) to 15.6(2)T, and update the vendor's LoC reflecting compliance with v.150.1.	Minor	CLOSED
Per the vendor's LoC, the SUT has No Audio Payload Type Requirements for SCIP-216 Compliant Gateways. This discrepancy was fixed and successfully tested with DTRs 7 and 11, which included the following updates: update UC Manager (UCM) from version 10.5.2.12901-1 to version 10.5.2 SU3, and update the 35xx series analog voice gateways, and the 29xx series and 39xx series ISR voice gateways from IOS 15.4(3) to 15.6(2)T, and update the vendor's LoC reflecting compliance with v.150.1.	Minor	CLOSED
The SUT is unable to perform "Incoming Trunk Preemption for Reuse of an Unanswered call" (Ringing @ SUT) via T1 CAS.	Minor	See note 1.
Cisco's ISR 44xx Trunk Side Media Gateway does not support v.150.1.	Minor	See note 1.
During testing, the SUT 39xx/29xx IWG/SBC routers caused one-way video failure when v.150.1 was enabled. This discrepancy was fixed and successfully tested with the update of the IOS IWG/SBC routers from 15.4(3)M5 to 15.5(3)M3 included with DTRs 7 and 11.	Minor	CLOSED
NOTES: 1. DISA has accepted the vendor's POA&M and has adjudicated this discrepancy as minor. 2. DISA has adjudicated this discrepancy as minor, with change requirement. 3. DISA has adjudicated this discrepancy as minor and stated the intent to remove this requirement from the UCR and apply it to a DoD STIG.		

Table 1. Conditions (continued)

LEGEND:

AES	Advanced Encryption System	OCSP	Online Certificate Status Protocol
AS	Application Server	POA&M	Plan of Action and Milestones
BPA	Blocked Precedence Announcement	ROEI	ROUTINE Only End Instrument
CAS	Channel Associated Signaling	RTS	Real Time Services
DG	Deployment Guide	SBC	Session Border Controller
DISA	Defense Information System Agency	SCIP	Secure Communications Interoperability Protocol
DoD	Department of Defense	SDP	Session Description Protocol
DTR	Desktop Review	SRTP	Secure Real-time Transport Protocol
EI	End Instrument	STIG	Security Technical Implementation Guide
GCM	Galois/Counter Mode	SU	Service Update
IOS	Internetwork Operating System	SUT	System Under Test
ISR	Integrated Services Router	T1	Digital Transmission Link Level 1
IWG	Interworking Gateway	UCM	Unified Communications Manager
LoC	Letters of Compliance	UCR	Unified Capabilities Requirements
LRDB	Local RTS Routing Database	VVoIP	Voice and Video over Internet Protocol
MRDB	Master RTS Routing Database		
NTPv3	Network Time Protocol version 3		

3. **Interoperability Status.** Table 2 provides the SUT interface interoperability status and Table 3 provides the Capability Requirements (CR) and Functional Requirements (FR) status. Table 4 provides the UC APL product summary.

Table 2. Interface Status

Interface (See note 1.)	Status	Remarks
Network Management Interfaces		
10BaseT (R)	Met	The SUT met the critical CRs and FRs for the IEEE 802.3i interface.
100BaseT (R)	Met	The SUT met the critical CRs and FRs for the IEEE 802.3u interface.
1000BaseT (C)	Met	The SUT met the critical CRs and FRs for the IEEE 802.3ab interface.
Network Interfaces (Line and Trunk)		
10BaseT (R)	Met	The SUT met the critical CRs and FRs for the IEEE 802.3i interface with the SUT PEIs and softphones.
100BaseT (R)	Met	The SUT met the critical CRs and FRs for the IEEE 802.3u interface with the SUT PEIs and softphones.
1000BaseT (R)	Met	The SUT met the critical CRs and FRs for the IEEE 802.3ab interface with the SUT PEIs and softphones.
2-wire analog (R)	Met	The SUT met the critical CRs and FRs for the 2-wire analog interface with the SUT 2-wire secure and non-secure analog instruments.
ISDN BRI (C)	Not Tested	The SUT offers this interface; however, it was not tested because it does not support Assured Services and is not required for an ESC.
Legacy Interfaces (External)		
10BaseT (C)	Met	The SUT met the critical CRs/FRs for IEEE 802.3i for the AS-SIP trunk.
100BaseT (C)	Met	The SUT met the critical CRs/FRs for IEEE 802.3u for the AS-SIP trunk.
1000BaseT (C)	Met	The SUT met the critical CRs/FRs for IEEE 802.3ab for the AS-SIP trunk.
ISDN T1 PRI (ANSI T1.619a) (R)	Met	The SUT met the critical CRs/FRs. This interface provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2 (R)	Met	The SUT met the critical CRs/FRs. This interface provides PSTN connectivity.
T1 CCS7 (ANSI T1.619a) (C)	Not Tested	The SUT does not support this conditional interface.

Table 2. Interface Status (continued)

Interface (See note 1.)	Status	Remarks
Legacy Interfaces (External) (continued)		
T1 CAS (C)	Partially Met (See note 2.)	The SUT partially met threshold CRs/FRs for DTMF. This interface provides legacy DSN connectivity.
E1 PRI (ITU-T Q.955.3) (C)	Met (See note 3.)	The SUT met the critical CRs/FRs. This interface provides OCONUS MLPP connectivity in ETSI-compliant countries.
E1 PRI (ITU-T Q.931) (C)	Met (See note 3.)	The SUT met the critical CRs/FRs. This interface provides OCONUS connectivity in ETSI-compliant countries.

NOTES:
1. The SUT high-level CR and FR ID numbers depicted in the Threshold CRs/FRs column can be cross-referenced in Table 3. These high-level CR/FR requirements refer to a detailed list of requirements provided in Reference (c), Enclosure 3.
2. The SUT met the requirements with the exceptions noted in Table 1. DISA accepted the vendors POA&M and adjudicated these exceptions as minor.
3. The E1 Interface ITU-T Q.955.3 and Q.931 protocols were met without testing, based on JITC analysis because there has been no effective change in the SUT Trunk-Side Media Gateways, with to the E1 PRI interfaces since these gateways were tested and certified as part of the ESC 8 certification as depicted in Reference (h).

LEGEND:

10BaseT	10 Mbps Ethernet
100BaseT	100 Mbps Ethernet
1000BaseT	1000 Mbps Ethernet
ANSI	American National Standards Institute
AS-SIP	Assured Services Session Initiation Protocol
BRI	Basic Rate Interface
C	Conditional
CAS	Channel Associated Signaling
CCS7	Common Channel Signaling Number 7
CR	Capability Requirement
DISA	Defense Information System Agency
DSN	Defense Switched Network
DTMF	Dual Tone Multi-Frequency
E1	European Basic Multiplex Rate (2.048 Mbps)
ESC	Enterprise Session Controller
ETSI	European Telecommunications Standards Institute
FR	Functional Requirement
ID	Identification
IEEE	Institute of Electrical and Electronics Engineers
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
JITC	Joint Interoperability Test Command
Mbps	Megabits per second
MLPP	Multi-Level Precedence and Preemption
NI-2	National ISDN Standard 2
OCONUS	Outside the Continental United States
PEI	Proprietary End Instrument
POA&M	Plan of Action and Milestones
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.931	Signaling Standard for ISDN
Q.955.3	ISDN Signaling Standard for E1 MLPP
R	Required
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.619a	SS7 and ISDN MLPP Signaling Standard for T1

Table 3. SUT Capability Requirements and Functional Requirements Status

CR/FR ID	UCR Requirement (High-Level) (See note 1.)	UCR 2013 Reference	Status
1	Voice Features and Capabilities (R)	2.2	Partially Met (See note 2.)
2	Assured Services Admission Control (R)	2.3	Met
3	Signaling Protocols (R)	2.4	Met
4	Registration and Authentication (R)	2.5	Met
5	SC and SS Failover and Recovery (R)	2.6	Met (See note 3)
6	Product Interface (R)	2.7	Met
7	Product Physical, Quality, and Environmental Factors (R)	2.8	Met
8	End Instruments (including tones and announcements) (R)	2.9	Partially Met (See note 2.)
9	Session Controller (R)	2.10	Met
10	AS-SIP Gateways (C)	2.11	Partially Met (See note 2.)
11	Enterprise UC Services (R)	2.12	Partially Met (See note 2.)
12	Call Connection Agent (R)	2.14	Met

Table 3. SUT Capability Requirements and Functional Requirements Status (continued)

CR/FR ID	UCR Requirement (High-Level) (See note 1.)	UCR 2013 Reference	Status
13	CCA Interaction with Network Appliances and Functions (R)	2.15	Met
14	Media Gateway (R)	2.16	Met
15	Worldwide Numbering & Dialing Plan (R)	2.18	Met
16	Management of Network Devices (R)	2.19	Partially Met (See note 2.)
17	v.150.1 Modem Relay Secure Phone Support (R)	2.20	Met (See note 4.)
18	Requirements for Supporting AS-SIP Based Ethernet Devices for Voicemail Systems (C)	2.21	Not Tested
19	Local Attendant Console Features (O)	2.22	Not Tested
20	MSC and SSC (O)	2.23	Not Tested (See note 5.)
21	MSC, SSC, and Dynamic ASAC Requirements in Support of Bandwidth-constrained links (O)	2.24	Not Tested (See note 5.)
22	Other UC Voice (R)	2.25	Partially Met (See note 2.)
23	Information Assurance Requirements (R)	4	Met (See note 6.)
24	IPv6 Requirements (R)	5	Met
25	AS-SIP 2013 (R)	AS-SIP	Partially Met (See note 2.)

NOTES:

1. The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Reference (c), Enclosure 3.
2. The SUT met the requirements with the exceptions noted in Table 1. DISA accepted the vendors POA&M and adjudicated these exceptions as minor.
3. The SUT met the requirements based on previous test results captured during the certification testing of the Cisco LSC UCM version 8.6.1. JITC analysis determined there has been no effective change in the code of the Cisco IWG as it relates to failover since it was tested with LSC UCM version 8.6.1.
4. This discrepancy was fixed and successfully tested with DTRs 7 and 11, which included the following updates: UCM from version 10.5.2.12901-1 to 10.5.2 SU3 and update the 29xx, 35xx and 39xx VG IOS from IOS 15.4(3) to 15.6(2)T and the SBC/IWG update from IOS 15.4(3)M5 to IOS 15.5(3)M3.
5. This optional requirement applies specifically to a LSC.
6. Security is tested by DISA-led Information Assurance test teams and the results published in a separate report, Reference (g).

LEGEND:

ASAC	Assured Services Admission Control	LSC	Local Session Controller
AS-SIP	Assured Services Session Initiation Protocol	MSC	Modular Services Card
C	Conditional	O	Optional
CCA	Call Connection Agent	POA&M	Plan of Action and Milestones
CR	Capability Requirement	R	Required
DISA	Defense Information System Agency	SBC	Session Border Controller
DTR	Desktop Review	SC	Session Controller
FR	Functional Requirement	SS	Softswitch
ID	Identification	SSC	Secured Services Client
IOS	Internetwork Operating System	SU	Service Update
IPv6	Internet Protocol version 6	SUT	System Under Test
IWG	Interworking Gateway	UC	Unified Capabilities
JITC	Joint Interoperability Test Command	UCM	Unified Communications Manager
LoC	Letters of Compliance	UCR	Unified Capabilities Requirements

Table 4. UC APL Product Summary

Product Identification	
Product Name	Cisco Enterprise Session Controller (ESC) 15
Software Release	10.5.2 SU3 (See note 1.)
UC Product Type(s)	Enterprise Session Controller (ESC) or Local Session Controller (LSC)
Product Description	ESC for Type 1, 2, and 3 Environments or as a LSC

Table 4. UC APL Product Summary (continued)

Product Components (See note 2.)	Component Name (See notes 3 and 4.)	Version	Remarks
Unified Communications Manager	**Cisco Unified Communications Manager**	10.5.2 SU3	See note 1.
Session Management Edition	**Cisco Session Management Edition**	10.5.2 SU3	See note 1.
Unified Communications Manager	**Cisco Unified Communications Manager**	10.5.2 SU3	See note 1.
Cisco Unity Connection	**Cisco Unity Connection**	10.5.2 SU3	See note 1.
Instant Messaging & Presence Server	**Instant Messaging & Presence Server**	10.5.2.22900-2	
Cisco WebEx Meetings Server	**Cisco WebEx Meetings Server**	2.5	
E911 management system	RedSky E911 Management System	6.3.1	See note 5.
Interworking Gateway	IWG on 2901 ISR G2, IWG on 2911 ISR G2, IWG on 2921 ISR G2, IWG on 2951 ISR G2, IWG on 3925 ISR G2, IWG on 3925E ISR G2, **IWG on 3945 ISR G2,** IWG on 3945E ISR G2	IOS 15.5(3)M3	See note 6.
Session Border Controller	SBC on 2901 ISR G2, SBC on 2911 ISR G2, SBC on 2921 ISR G2, SBC on 2951 ISR G2, SBC on 3925 ISR G2, SBC on 3925E ISR G2, **SBC on 3945 ISR G2,** SBC on 3945E ISR G2	IOS 15.5(3)M3	See note 6.
Voice Gateway	2901 ISR G2, 2911 ISR G2, 2921 ISR G2, **2951 ISR G2**, 3925 ISR G2, 3925E ISR G2, **3945 ISR G2,** 3945E ISR G2	IOS 15.6(2)T	See note 7.
Interworking Gateway/Session Border Controller	IWG/SBC on 2901 ISR G2, IWG/SBC on 2911 ISR G2, IWG/SBC on 2921 ISR G2, IWG/SBC on **2951 ISR G2**, IWG/SBC on 3925 ISR G2, IWG/SBC on 3925E ISR G2, **IWG/SBC on 3945 ISR G2,** IWG/SBC on 3945E ISR G2	IOS 15.5(3)M3	See note 6.
Voice Gateway	4321 ISR G3, 4331 ISR G3, 4351 ISR G3, 4431 ISR G3, **4451-X ISR G3**	IOS-XE 3.15	See note 8.
Analog Voice Gateway	**VG202XM** and VG204XM Analog Voice Gateway	IOS 15.4(3)M4	See note 9.
Analog Voice Gateway	**VG350,** VG310 and VG320 Analog Voice Gateway	IOS 15.6(2)T	See note 7.
Jabber (Voice and Video Soft Client)	Cisco Jabber for Windows	Version 11.0 Windows 7	See note 10.
IP Phone (Voice)	IP Phone 6901	9.4.1.3	See note 11.
IP Phone (Voice)	IP Phone 6911	9.4.1.3	See note 11.
IP Phone (Voice)	IP Phone 6921	9.4.1.3	See note 11.
IP Phone (Voice)	IP Phone 6941	9.4.1.3	See note 11.
IP Phone (Voice)	IP Phone 6945	9.4.1.3	See note 11.
IP Phone (Voice)	**IP Phone 6961**	9.4.1.3	See note 11.
IP Phone (Voice)	IP Phone 7811	10.3.1	See note 12.
IP Phone (Voice)	IP Phone 7821	10.3.1	See note 12.
IP Phone (Voice)	IP Phone 7841	10.3.1	See note 12.
IP Phone (Voice)	**IP Phone 7861**	10.3.1	See note 12.
IP Phone (Voice)	IP Phone 7906G	9.4.2	See note 11.
IP Phone (Voice)	IP Phone 7911G	9.4.2	See note 11.
IP Phone (Voice)	Unified IP Phone 7931G	9.4.2	See note 11.
Wireless IP Phone	IP Phone 7925G	1.4.1	See note 13.
IP Phone (Voice)	Unified IP Phone 7941G	9.4.2	See note 11.
IP Phone (Voice)	Unified IP Phone 7941G-GE	9.4.2	See note 11.
IP Phone (Voice)	Unified IP Phone 7942G	9.4.2	See note 11.
IP Phone (Voice)	Unified IP Phone 7945G	9.4.2	See note 11.
IP Phone (Voice)	Unified IP Phone 7961G	9.4.2	See note 11.
IP Phone (Voice)	Unified IP Phone 7961G-GE	9.4.2	See note 11.
IP Phone (Voice)	Unified IP Phone 7962G	9.4.2	See note 11.
IP Phone (Voice)	CIS DTD-7962-T2 and CIS DTD-7962-TSG-01	9.4.2	See notes 11 and 14.

Table 4. UC APL Product Summary (continued)

Product Components (See note 2.)	Component Name (See notes 3 and 4.)	Version	Remarks
IP Phone (Voice)	API DNT502-xx	9.4.2	See notes 11 and 15.
IP Phone (Voice)	Unified IP Phone 7965G	9.4.2	See note 11.
IP Phone (Voice)	CIS DTD-7965-TSGB	9.4.2	See notes 11 and 16.
IP Phone (Voice)	API DNC503-xx	9.4.2	See notes 11 and 17.
IP Phone (Voice)	Unified IP Phone 7970G	9.4.2	See note 11.
IP Phone (Voice)	Unified IP Phone 7971G	9.4.2	See note 11.
IP Phone (Voice)	**Unified IP Phone 7975G**	9.4.2	See note 11.
IP Phone (Voice)	CIS DTD-7975-T2	9.4.2	See notes 11 and 18.
IP Phone (Voice and Video)	**Unified IP Phone 9951**	9.4.2	See note 12.
IP Phone (Voice and Video)	**Unified IP Phone 9971**	9.4.2	See note 12.
IP Phone (Voice and Video)	**Unified IP Phone 8811**	10.3.1	See notes 12 and 19.
IP Phone (Voice and Video)	Unified IP Phone 8831 Conference Phone	10.3.1	See notes 12 and 19.
IP Phone (Voice and Video)	API EM-8831-xx	10.3.1	See notes 12, 19 and 20.
IP Phone (Voice and Video)	Unified IP Phone 8851 and 8851NR	10.3.1	See notes 12 and 19.
IP Phone (Voice and Video)	API EM-8851–xx	10.3.1	See notes 12, 19 and 21.
IP Phone (Voice and Video)	CIS DTD-8851-01	10.3.1	See notes 12 and 22.
IP Phone (Voice and Video)	**Unified IP Phone 8841**	10.3.1	See notes 12 and 19.
IP Phone (Voice and Video)	API EM-8841–xx	10.3.1	See notes 12, 19 and 23.
IP Phone (Voice and Video)	API EL1-8841–xx	10.3.1	See notes 12, 19 and 24.
IP Phone (Voice and Video)	CIS DTD-8841-01	10.3.1	See notes12 and 25.
IP Phone (Voice and Video)	CIS DTD-8841T-01-L1	10.3.1	See notes 12 and 25.
IP Phone (Voice and Video)	CIS DTD-8841-02	10.3.1	See notes 12 and 25.
IP Phone (Voice and Video)	**Unified IP Phone 8845**	10.3.2	See notes 12 and 19.
IP Phone (Voice and Video)	Unified IP Phone 8861	10.3.1	See notes 12 and 19.
IP Phone (Voice and Video)	**Unified IP Phone 8865**	10.3.2	See notes 12 and 19.
IP Phone (Voice)	**Unified IP Phone 8961**	9.4.2	See note 26.
Expansion Module	Unified IP Phone Expansion Module7915	Not Applicable	
Expansion Module	Unified IP Phone Expansion Module7916	Not Applicable	
Expansion Module	Unified IP Phone Color Expansion Module 8900, 9900 models	Not Applicable	
Expansion Module	Unified IP Phone KEM Expansion module for 8800 series IP Phones	Not Applicable	
Video Teleconference (Voice and Video)	**Cisco DX70**, Cisco DX80, **Cisco DX650**	10.2.5	See notes 12 and 27.
Video Teleconference (Video)	**SX-10**, **SX-20**, **SX-80**	CE 8.2	See notes 28 and 29.

Table 4. UC APL Product Summary (continued)

Product Components (See note 2.)	Component Name (See notes 3 and 4.)	Version	Remarks
Video Teleconference	MX200, MX300, VX-Tactical, and VX-Clinical Assistant	TC 7.3	See notes 28 and 30.
Video Teleconference	MX200G2 and MX300G2	CE 8.2	See notes 28, 29, and 30
Video Teleconference	MX700, MX800, and MX800 Dual	CE 8.2	See notes 28, 29, 31, and 32.
Video Teleconference	**Cisco TelePresence VCS**	X8.8	See note 33.
Softphone	Cisco IP Communicator	8.6.1.0	See note 34.
Multipoint Control Unit	**Cisco Meeting Server**	2.0	See note 35.
Analog PSTN mode	GD vIPer	5.0.3	See note 36.
SCCP mode	GD IP vIPer	5.1.0	See note 36.
Firewall and proxy	ASA 55xx	9.4.2	See note 37.
Secure mobility VPN client	AnyConnect	4.3	See note 38.

NOTES:
1. The UCM version was updated from 10.5.2.12901-1 to 10.5.2 SU3 with DTR 7.
2. The detailed component and subcomponent list is provided in Reference (c), Enclosure 3.
3. Components bolded and underlined were tested by JITC. The other components in the family series were not tested but are also certified for joint use. JITC certifies those additional components because they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes.
4. A comprehensive list of supported hardware configurations can be found by selecting the "Cisco Unified Communications on the Cisco Unified Computing System" link at the following URL: www.cisco.com/go/swonly.
5. The SUT is certified with any RedSky E911 Management system version listed on the UC APL and certified with the Cisco UCM. The RedSky E911 Management System is purchased separate from the SUT. E911 management is only required for an ESC. The RedSky E911 management system was not tested with the SUT, but was determined by JITC analysis to be compliant to E911 management requirements for an ESC based upon previous test data collected on the same hardware platform with similar software, that did not change the management functionality when it was updated to release 10.5.2.
6. The SUT SBC/IWGs were tested and are certified with Cisco IOS Release 15.4(3.0h)M4, which is the prerelease build for Cisco IOS Release 15.4(3)M5. The SUT SBC/IWGs were updated from IOS Release 15.4(3.0h)M4 to IOS Release 15.4(3)M5 with DTR 6. The SUT SBC/IWGs were updated from IOS Release 15.4(3)M5 to IOS Release 15.5(3)M3 with DTRs 7 and 11 V&V.
7. The VG35xx series analog voice gateways, the 29xx series and 39xx series ISR voice gateways were updated from IOS 15.4(3) to 15.6(2)T with DTR 7.
8. The 44xx ISR MG will not be updated to include v.150.1 functionality.
9. The VG202XM and VG204XM with release IOS 15.4(3)M4 were included with DTR 2. Interoperability testing was conducted on the VG202XM 2-port analog voice gateway. The VG204XM gateway uses same software and effectively is the same as the VG202XM except it supports four (4) RJ-11 ports instead of two (2). Based on this difference, JITC determined that the VG204XM functions identically to the VG202XM for interoperability certification purposes and therefore is also covered in this certification. The VG202XM and VG204XM do not have the capability to support recovered timing and as a result they do not support optional secure calls (e.g., v.150.1), but they do support non-secure voice and fax calls. Only the VG3xx series support recovered timing and therefore, support secure calls.
10. Jabber Video and Voice soft clients support SIP protocol only and are certified as Routine only end instruments. The Jabber collaboration soft client version 11.0 for Windows 7 for use over a VPN was included with DTR 12.
11. These IP phones support both SCCP and SIP protocol, however only SCCP was tested and is certified for assured services MLPP.
12. These IP phones support SIP protocol only and are certified for assured services MLPP.
13. The Cisco CP-7925G Wireless phone was added to this certification as an approved EI that supports SCCP for signaling based upon JITC analysis. The analysis is based on no change in the software or hardware since this wireless phone was previously tested with Cisco UCM Release 8.6.1 (20010-5) as a Local Session Controller under Tracking Number 1108301 and the fact that the phone design is based on the 79xx series IP phone, which fully demonstrated compliance to the End Instrument requirements with the SUT.
14. JITC analysis determined the CIS DTD-7962-T2 and CIS DTD-7962-TSG-01 are identical in regards to interoperability and IA posture to the 7962G IP Phone and were added to the list of certified EIs with DTR 18.
15. JITC analysis determined the API DNT502-xx is identical in regards to interoperability and IA posture to the 7962G IP Phone and was added to the list of certified EIs with DTR 10.
16. JITC analysis determined the CIS DTD-7965-TSGB is identical in regards to interoperability and IA posture to the 7965G IP Phone and was added to the list of certified EIs with DTR 18.
17. JITC analysis determined the API DNC503-xx is identical in regards to interoperability and IA posture to the 7965G IP Phone and was added to the list of certified EIs with DTR 10.
18. JITC analysis determined the CIS DTD-7975-T2 is identical in regards to interoperability and IA posture to the 7975G IP Phone and was added to the list of certified EIs with DTR 18.

Table 4. UC APL Product Summary (continued)

NOTES (continued):

19. The Remote Client Gateway functionality with Cisco AnyConnect VPN was added to the following SUT components: 88XX VoIP EIs, 8845 and 8865 VVoIP EIs, and the DX series VVoIP EIs, included with DTR 8.
20. JITC analysis determined the API EM-8831-xx is identical in regards to interoperability and IA posture to the 8831 IP Phone and was added to the list of certified EIs with DTR 10.
21. JITC analysis determined the API EM-8851-xx is identical in regards to interoperability and IA posture to the 8851 IP Phone and was added to the list of certified EIs with DTR 10.
22. JITC analysis determined the CIS DTD-8851-01 is identical in regards to interoperability and IA posture to the 8851 IP Phone and was added to the list of certified EIs with DTR 18.
23. JITC analysis determined the API EM-8841-xx is identical in regards to interoperability and IA posture to the 8841 IP Phone and was added to the list of certified EIs with DTR 10.
24. JITC analysis determined the API EL1-8841–xx is identical in regards to interoperability and IA posture to the 8841 IP Phone and was added to the list of certified EIs with DTR 10.
25. JITC analysis determined the CIS DTD-8841-01 Release 10.3.1, CIS DTD-8841-01-L1 Release 10.3.1 and the CIS DTD-8841-02 Fiber Enabled Release 10.3.1 are identical in regards to interoperability and IA posture to the 8841 IP Phone and were added to the list of certified EIs with DTR 18.
26. These IP phones support SCCP and SIP protocol, however only SIP was tested and is certified with assured services MLPP.
27. The SUT DX series VVoIP devices fail to establish two-way video when calls are placed to a Polycom Group Series video EI. This discrepancy was fixed with the updates included in DTRs 7 and 11.
28. These IP phones support SIP protocol only and are certified for ROUTINE Only.
29. The SX10, SX20, and SX80 video teleconference codec software, and the MX200G2, MX300G2, MX700, MX800 Dual, and MX800 codec software were updated from version TC 7.3 to version CE 8.1.0 with DTR 5. The SX10, SX20 and SX80 video teleconference codec software, and the MX200G2, MX300G2, MX700, MX800 Dual, and MX800 codec software were updated from version CE 8.1.0 to version CE 8.2 with DTR 13.
30. These Video Teleconference End Instruments were not tested and are certified based on similarity to the SX-20.
31. These Video Teleconference End Instruments were not tested and are certified based on similarity to the SX-80.
32. The MX800 Dual with release TC 7.3 was included with DTR 4. The Cisco MX800 Dual uses the same hardware and software as the MX800 MLPP video phone, except it supports two monitors, instead of just one.
33. The Cisco TelePresence VCS release X8.8 was added to support transcoding for video H.323 end instruments to SIP and vice versa in DTR 15.
34. The Cisco IP Communicator with release 8.6.1.0 was included with DTR 1. The Cisco IP Communicator release 8.6.1.0 was previously tested and certified with the Cisco UCM LSC release 8.6.1 under UCCO tracking number 1108301. JITC analysis determined that there has been no change in the IP Communicator hardware or software since that LSC UCM testing and that the Cisco UCM 8.6.1 and the Cisco UCM 10.5.2 have very similar performance characteristics and are built off of the same baseline code. Therefore, JITC added the IP Communicator to this certification letter without testing.
35. The Acano video conferencing system with release 1.8.10 was included with DTR 3. DTR 16 upgraded the software from release 1.8.10 to 2.0, as well as rebrands the existing MCU from the Acano Server to the CMS.
36. The the secure GD vIPer PSTN Mode version 5.0.3, and the secure GD vIPer IP SCCP mode version 5.1.0 were added to the SUT with DTR 7.
37. The SUT Firewall and Proxy ASA 55xx version 9.4.2 was included with DTR 8.
38. The SUT secure mobility Cisco AnyConnect VPN client version 4.3 was included with DTR 8.

LEGEND:

API	Application Programming Interface	MCU	Multipoint Conference Unit
APL	Approved Product List	MG	Media Gateway
ASA	Automatic Security Authentication	MLPP	Multilevel Precedence and Preemption
CE	Collaboration Endpoint	PSTN	Public Switched Telephone Network
CMS	Cisco Meeting Server	SBC	Session Border Controller
CP	Conference Phone	SCCP	Skinny Client Control Protocol
DTR	Desktop Review	SIP	Session Initiation Protocol
EI	End Instrument	SU	System Update
EM	Extension Mobility	SUT	System Under Test
ESC	Enterprise Session Controller	TC	Tanberg Communicator
G2	Generation 2	UC	Unified Capabilities
G3	Generation 3	UCCO	Unified Capabilities Certification Office
GD	General Dynamics	UCM	Unified Communications Manager
IA	Information Assurance	URL	Uniform Resource Locater
IOS	Internetwork Operating System	V&V	Verification and Validation
IP	Internet Protocol	VCS	Video Communication Server
ISR	Integrated Services Router	VG	Voice Gateway
IWG	Interworking Gateway	VPN	Virtual Private Network
JITC	Joint Interoperability Test Command	VoIP	Voice over IP
KEM	Key Extension Module	VVoIP	Voice/Video over IP
LSC	Local Session Controller		

4. **Test Details.** The extension of this certification is based upon DTR 13. The original certification, documented in Reference (c), is based on interoperability testing, Defense Information System Agency (DISA) adjudication of open test discrepancy reports (TDRs), review of the vendor's Letters of Compliance (LoC), and DISA Certifying Authority (CA) Recommendation for inclusion on the UC APL. Voice over Internet Protocol (VoIP) System Acceptance Testing (SAT) was conducted on an operational Cisco ESC with software release 10.5 by Network Enterprise Technology Command (NETCOM) during May, June, and July of 2015 documented in Reference (d). Limited data (Call Pickup, Voicemail and Dual Tone Multi-Frequency [DTMF] recognition) from the SAT was included in this certification. Testing was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 9 November through 7 December 2015 using test procedures derived from References (e) and (f). Review of the vendor's LoC was completed on 14 December 2015. DISA adjudication of outstanding TDRs was completed on 4 December 2015. Information Assurance (IA) testing was conducted by DISA-led IA test teams and the results are published in a separate report, Reference (g).

This DTR 13 was requested to update the SX10, SX20, and SX80 series, video teleconference codec software, as well as update the MX200G2, MX300G2, MX700, MX800 Dual, and MX800 codec software from version CE 8.1.0 to version CE 8.2. JITC analysis determined that the upgrades to the telepresence codecs will have no impact on the interoperability certified features and functions of the SUT, therefore no interoperability testing was required. Additionally, DISA has approved the IA posture of the SUT for this DTR based on IA testing conducted by JITC-led IA test teams and the results are published in a separate report, Reference (g). Therefore, JITC approves this DTR.

5. **Additional Information.** JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Sensitive but Unclassified IP Data (formerly known as NIPRNet) e-mail. Interoperability status information is available via the JITC System Tracking Program (STP). STP is accessible by .mil/.gov users at https://stp.fhu.disa.mil/. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at https://jit.fhu.disa.mil/. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly from the UCCO, e-mail: disa.meade.ie.list.unified-capabilities-certification-office@mail.mil. All associated information is available on the DISA UCCO website located at http://www.disa.mil/Services/Network-Services/UCCO.

6. **Point of Contact (POC).** The JITC point of contact is Mr. Joseph Schulte, commercial telephone (520) 538-5100, DSN telephone 879-5100, FAX DSN 879-4347; e-mail address joseph.t.schulte.civ@mail.mil; mailing address Joint Interoperability Test Command, ATTN: JTE (Mr. Joseph Schulte) P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The UCCO tracking number for the SUT is 1525401.

FOR THE COMMANDER:

CLARK.BRADLEY.ALAN.1231466564

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Date: 2016.10.06 08:14:00 -07'00'

Enclosure a/s

For RIC HARRISON
Chief
Networks/Communications and UC Division

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UCCO

ADDITIONAL REFERENCES

(c) Joint Interoperability Test Command, Memo, JTE, "Joint Interoperability Certification of the Cisco Enterprise Session Controller (ESC) 15 Release 10.5.2," 17 December 2015

(d) Network Enterprise Technology Command (NETCOM) Fort Huachuca "Voice over Internet Protocol (VoIP) Systems Acceptance Test version 1.3," Draft

(e) Joint Interoperability Test Command, "Enterprise Session Controller (ESC) Test Procedures for Unified Capabilities Requirements (UCR) 2013, Errata 1," 2 July 2015

(f) Joint Interoperability Test Command, "Session Controller (SC) Test Procedures for Unified Capabilities Requirements (UCR) 2013," 16 October 2015

(g) Joint Interoperability Test Command, "Information Assurance (IA) Findings Summary For Cisco Enterprise Session Controller (ESC) 15 Release (Rel.) 10.5 (Tracking Number 1525401)," September 2016

(h) Joint Interoperability Test Command, Memo, JTE, "Joint Interoperability Certification of the Cisco Enterprise Session Controller (ESC) 8," 13 June 2014